

LP600N

SIP IP Phone for IP-PBX Application



Grey



Support 3.5mm plug-in Ear Phone and Microphone.



Blue



2-Line and 3 programmable



Function
Keys

Support 2 Simultaneous Calls at one IP-PBX Server account

Rich Telephony Features

Plug and Play to work with SIPPBX 6200x IP-PBX Server

HTTP Provision provide auto configuration

Intelligent Phone book Name dialing

Support Multi-Party Voice Conference

Color : Blue, Black and Grey

LP600N is a SIP IP Phone which work as extension of Welltech SIPPBX 6200x series (An IP-PBX server, for instance, SIPPBX6200A, SIPPBX6200S, SIPPBX6200GS, SIPPBX6200N and ePBX100A-128). LP600N IP Phone needs to configure those information includes Line number, Phone Book, IP-PBX or SIP Server, dial Plans and others refer to parameters to link with IP-PBX which can be configured by SIPPBX6200's Administrator to fulfill without configuring LP600N one by one. The firmware upgraded to unique LP600N can be managed by SIPPBX6200x Administrator automatically. This convenient feature gives IP-PBX manager to configure office IP-PBX and LP600N features easily and effectively.

LP600N

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Benefits

Support 2 Simultaneous Calls

LP600N Supports one Line number to register to one SIPPBX 6200x IP-PBX server and is able to make 2 phone calls. By using rich telephony feature, these IP-PBX features are available to increase your jobs efficiency.

Plug & Play with SIPPBX 6200 IP-PBX Series

LP600N related configurable features were stored at SIPPBX6200x IP-PBX server according to its MAC address before installation at customer site. This behavior gives office IP-PBX manager and user has enough time to verify his/her required feature and group planning to suit personnel demand before installation. Once LP600N installs at local site and link to DHCP server and SIPPBX6200x IP-PBX server automatically, it accepts to download its pre-configured personnel parameters and phone number as well. In a minute, LP600N is ready to use according to personnel configuration demand. LP600N's installation is so easy and support Plug & Play without consuming a long installation time one by one. All those personnel information are stored at SIPPBX6200x IP-PBX server and can be exported to make a backup in case of hardware failure at SIPPBX 6200x or LP600N in order to resume your original configuration.

Rich Telephony Features

LP600N was designed to work with SIPPBX6200x IP-PBX and ePBX100A-128 (30 users small IP-PBX) in order to provide rich telephony PBX feature. For instance, Call Hold, Call Pick-up, Multi-party Conference, Call Transfer, Call Forwarding, Do Not Disturb, Phone book, Mute, Voice Mail, Headset, Missed calls, call records, use your preference Music as Ring and distinctive ring as well.

Quick Phone Book name Dialing

By using intelligent phone book name or phone number search engine inside, LP600N IP Phone user can dial number from their phone book easily by using the navigation keypad without using the difficult English character input from keypad.

Support Up to 8 voice Parties Conference Calls

LP600N itself supports voice mixer for 3-Way Voice Conference Calls. Besides, SIPPBX6200x IP-PBX server provides 8-Party Voice Conference Bridge. LP600N Switch Conference call Bridge to SIPPBX6200x Server automatically when it requires more than 3-Party conference calls.

SPECIFICATION

Interface:

Ethernet port (RJ-45, 10/100 base-T)

1-LAN port, for connecting to switch

1-PC port for connecting to PC

10/100 based Switch

PoE (IEEE 802.3af) at LAN port : **LP600NA only**

Earphone/Microphone Jack (3.5mm) for Headset

Handset Jack (RJ-10)

DC 12V power input Jack

LCD Display:

Display Format: 16 Characters (W) x 2 lines (H)

View Size: 64(W) x 17.9 (H) mm

LCD Type: TN

LCD Language Option: **English, Chinese**

IP Network connection

IPv4 (RFC 791),

MAC Address (IEEE 802.3)

Static IP

DHCP Client (RFC 2131)

PPPoE

Provide two DNS Server IP Address

TCP/UDP (RFC 793/768)

RTP/RTCP (RFC 1889/1890)

IPv4 ICMP (RFC 792)

TFTP Client

VoIP VLAN Support (802.1Q/802.1P)

VLAN ID Range : 2 to 4096

VLAN Priority : 0 to 7 (highest priority)

HTTP/HTTPS Server

QoS Support

SIP Protocol :

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RFC3261 compliance
Support 1 Line number, 2 calls at one SIP Register Account
Support Primary and Backup SIP Proxy
SIP Account Registration: Active, Auto Provision or Manual
 configure provision server.
SIP Transport Type: UDP, TCP, TLS
NAT Keep Alive Time
SIP UDP Protocol
Configurable SIP Local UDP, TCP and TLS port
SIP QoS Type : DiffServ and TOS
Voice RTP QoS Type : DiffServ and TOS
Configurable Voice RTP port
SIP Hold type
Support SIP compact Form
SIP Session Timer (RFC 4028)
SIP Timer
MD5 Digest Authentication
Reliability of provisional responses PRACK (RFC3262)
Early/Delay media support
Offer/answer (RFC3264)
Message Waiting Indication (RFC3842)
Event Notification (RFC3265)
REFER (RFC3515)
Support DNS SRV to locate SIP Server (RFC 3263)
Support STUN NAT Traversal
Support "rport" parameter (RFC 3581)

Audio Codec :

G.711 A-law/ μ -law, G.723.1 (6.3K/5.3K)
G.729A, GSM 6.10 (full rate)
Voice Codec Priority decision site : Local or Remote
Voice Codec Payload Size (ms) configuration
Silence Suppression
VAD/CNG
Adaptive/Configure Jitter Buffer
AEC Tail Length (ms) configure
Automatic Gain Control

Preference Setting :

Customized Idle Text display name at LCD
Phone Book with desired incoming call Ring Tone or Music
Intelligent Phone Book name Dialing
Clock, Day Light Saving, Call-Timer
Call History of Missed, Received and Dialed number
Dial Plans
Digit Manipulation (DM)
Selectable Call Progress Tone
Personal Music Ring
Support Silence Ring
Auto Answer Mode

Call Features :

1 Line number under at one IP-PBX Server or SIP Server
Caller ID display or inhibit
Voice mail with Indication
Speed Dialing
Call Waiting/Switching between Calls
Call Forward: Busy, Unconditional, No Answer

Block Anonymous Call
MIC and Volume configurable: Headset, Speaker, Handset, Ring
In-band/out of band DTMF (RFC 4733 (RFC 2833)/SIP INFO)
Configure RFC 2833 DTMF Payload Type
Voice and SIP Command Encryption
Send "REFER" without Hold
Command 180 W/O SDP after RTP : Play Remote Voice or
 Play Ring Back Tone
Program On-Net Call Telephone digits length
Send DTMF before connect
Program DTMF ON Time
Do not disturb
Call Hold
Call Mute
Call Transfer
Call Forward : Busy, Unconditional, No Answer
Block Incoming List phone number
Music-on-hold support (via IPPBX6200x or local)
3-Way Conference (over phone)
Server (IP-PBX 6200x) Conference Prefix code
Multi-parties conference (via IPPBX 6200x)
Distinct Ring between on-net and off-net calls (compatible with
 SIPPBX6200x)
Call Pickup (via IPPBX 6200x)
Call Park/Retrieve (via IPPBX 6200x)
Voice Broadcasting (via IPPBX6200x)
Barge-in & Barge-in Allowance List
Voice an SIP Encryption
Redialing/pre-dialing
Hot Line : Dial pre-defined number immediately or manual dial
 within desired due time (second)
Disable or Enable all features keys
3 User defined Keys to PSTN Line, Extension, Speed Dial or Speed
 dial with Input Text
Inter Digit Time Out : 1 to 10 seconds
Dial rule: Match dial prefix or Maximum digit Length
Digit Manipulation (DM):
 Matched Prefix code
 Start digit Position
 Stop digit Position
 Replaced number

MANAGEMENT :

SNTP time server with Daylight Saving
Variable Day, Month and Year display format
HTTP/HTTPS and Telnet Command
Enable or Disable HTTPS or Telnet Command
Configurable port number of HTTPS and Telnet
Multilingual Web User Interface
Administrative Telnet CLI
3 Levels of User Access Right with Password protection and
desired Web Language.
 Administrator
 Supervisor
 User
Built-in Rich Debug feature
Debug Phone Manager : Device Control, Call Control, Data Base
Debug Phone level: Emergency, Alert, Critical, Error, Warning,

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Notice, Information, Debug.
 Debug SIP Manager: Register, SIP Message, Other
 Debug DSP
 SYSLOG Server to collect Debug messages
 Support HTTP Provision from SIPPBX 6200x and ePBX100A-128
 IP-PBX Server
 LCD Administration Login
 Provides System Status
 Diagnostics (debug through syslog)
 Configuration Backup/Restore
 Reset to Default
 Dual Firmware Image Backup
 Auto provision by SIPPBX 6200x with Plug & Play
 Support HTTP provision through MAC address

Central Management by SIPPBX 6200x IP-PBX Server:

VLAN and DHCP Server provided by SIPPBX 6200x
 Extension Settings based on Mac address (up-to 3 Mac)
 Plug & Play without any settings on LP600N IP Phone
 Device-wide parameters
 Firmware Upgrade
 Time Settings
 Dial Plans

Service Code
 Office/Personal Phone Book

Environmental :

Actual Dimension: 20 × 9.5 × 22.3 CM (Desktop)
 Weight: 1.1kg (with packing)
 Operating Temp. & Humidity
 - Temp.: 0°C~45°C (32°F~113°F)
 - Humidity: 10%~90% relative humidity, non-condensing
 Power Adaptor:
 - INPUT: AC100V~240V, 50/60Hz
 - OUTPUT: DC 12V, 1.0A

Power Consumption of PoE : 4 Watts

Approvals:

CE, FCC, LVD and RoHS

Country of origin:

Made in Taiwan

Packing Accessories

LP600N IP Phone x 1 pcs
 AC to DC12V Power adaptor x 1 pcs
 CD User Manual

Warranty

One year

Ordering Information:

	LP600N	LP600NA
PoE	NO	YES
LCD Language	English	English
IPv4 ONLY	YES	YES
Headset, Ear Phone Microphone, Hand-Free	YES	YES
Wall Mount	NO	NO
Delivery Status	NOW	NOW

We provide a variety of IP Phone models which differ at features and application.

Features Comparison Table of IP Phone:

	LP801	LP389	LP600N	LP803	WP589	LP388	LP399
application	ITSP	ITSP	IPPBX	ITSP	ITSP	IPPBX,ITSP	ITSP
Line No.	2	3	2	3	1	2	3
Register to SIP Server	2	3	1	3	1	1	3
IPv6	NO	YES	NO	NO	NO	NO	NO
LCD	128x64 pixels	2x16 digits	2x16 digits	2x16 digits	1.44"Color	2x16 digits	2x16 digits
Handset	YES	YES	YES	NO	YES	YES	YES
Headset	YES, 3.5mm	YES, 3.5mm	YES, 3.5mm	YES, 3.5mm	NO	YES, RJ-9	NO
Speaker Phone	YES	YES	YES	YES	YES	YES	YES
WiFi Phone	NO	NO	NO	NO	YES	NO	NO
SIP Protocol	YES	YES	YES	YES	YES	YES	YES
Provision	TR069, HTTP	HTTP	IPPBX6200 built-in HTTP	YES	NO	NO	YES
Wall mount	NO	NO	NO	YES	Hand held	NO	NO
PoE Option	YES	YES	YES	YES	NO	YES	NO